METHOD FOR PACKET TRANSMISSION OF MULTIMEDIA DATA IN A NETWORK

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5 ## BACKGROUND OF THE INVENTION

Field of the Invention

The invention relates in general to a data transmission method, and more particularly to a method for transmitting multimedia data packets in a network.

Description of the Related Art

10   Nowadays, transmitting multimedia information combining text, audio, picture, and video via a network is a very common network usage.   Network users is expected to enjoy more and more multimedia services using the network as media. For instance, by using a "video on demand" (VOD) service, people can watch their favorite video or multimedia information immediately at any time by downloading via

15   the network.   Thus, people will not need to go to the cinema, or to go to the video shop for renting a video tape, or video compact disk for watching at home.   It is thanks to the development of network technology that brings people more efficient, more convenient, and better quality services.

Referring to FIG. 1, it illustrates a conventional network transmission system.

20   A transmission end 102 first dissembles data to be transmitted, denoted as data_in, into a set of packets 103.   An encoder 104 of transmission end 102 then encodes the

set of packets 103.   After that, the encoded packets 103 are sent to a router 108 in a

network 106.   On receiving packets 105, router 108 obtains the destination of the

packets, determines a route for corresponding packets 109, and sends them to the

destination, a receiving end 110.   Finally, the required data are obtained at receiving

5   end 110 by using a decoder 110 of receiving end 110 to decode packets 111.

Since the bandwidth of the network is fixed, only a maximum number of data

are allowed to be transmitted.   If there are too many data transmitting over the

network, congestion occurs.   When serious congestion occurs so that the router has

to receive a large number of packets at a time in excess of the number of packets that

10   the router can handle, bursty loss occurs that the router is to discard some received

packets randomly so as to lessen its loading.   Thus, the receiving end cannot receive

the data sent from the transmission end completely.   In the case of discarding packets

containing relatively unimportant information of the whole delivered information, the

influence on the reading and using of the whole delivered information is insignificant.

15   However, in the case of discarding packets containing important portion of the whole

delivered information, the influence on the reading and using of the whole delivered

information is significant.   In this way, the quality of service (QoS) is reduced.

In order to reduce the influence of bursty loss due to network congestion on

the quality of service, error recovery is applied.   There are two conventional

20   approaches to error recovery.   One is automatic repeat request (ARQ) and the other

is forward error correction (FEC).   In ARQ, when a receiving end receives a packet

from a transmission end, the receiving end sends an acknowledge message to the

transmission end.   After receiving the acknowledgment message from the receiving

end, the transmission end sends the next packet.   If the transmission end does not

receive the acknowledgment message from the receiving end within a predetermined

time period, it indicates that the packet could be discarded by the router and thus was

not delivered to the receiving end.   Thus, the transmission end re-transmits the

packet so that the receiving end can receive the packet correctly.   In FEC, a number

5   of redundancy packets are added to each packet to be sent.   In addition, if the

number of packets which are discarded by the router due to network congestion does

not exceed the number of the packets added to each packet, the discarded packets are

recoverable at the receiving end.

When ARQ approach is applied, a predetermined time is set at the transmitting

10   end, and the transmitting end waits for the response of the receiving end within the

predetermined time.   If the predetermined time is insufficiently short, it may occur

that the transmitting end does not receive the response packet after waiting for the

predetermined time, and the packet, in fact, is not discarded and the response packet is

delayed due to network congestion.   Because of the misinterpretation, the

15   transmission end re-transmits the packet that has been received by the receiving end.

In this way, the network traffic is increased and the loading of the router is increased.

On the other hand, if the predetermined time is too long, the time between succeeding

transmitted packets is long as well.   For transmitting real-time multimedia

information, if the time between succeeding transmitted packets is long, the

20   multimedia information cannot be played at the receiving end in a real-time manner.

Thus, it affects the quality of service of the multimedia information provided by the

network.   Further, the error detection time for re-transmitting packets are related to

the predetermined time and the degree of network congestion which cannot be

predicted such that the predetermined time is difficult to be determined.   Moreover, it

25   may occur that the packet sent by the transmission end is not discarded but the

response packet sent by the receiving end is discarded by the router.   In this case, the

transmission end re-transmits the packet due to the misinterpretation, and hence the

network bandwidth is wasted on retransmission, and the loading of the router is

increased.

5   If FEC approach is applied, the disadvantage of ARQ approach does not occur

since FEC approach is not to re-transmit packets for resolving the problem of burty

loss, but to add a number of redundancy packets to each packet.   In addition, the

more important the information is, the more the redundancy packets are added to the

packets of the information.   In this way, data recovery of the packets becomes more

10   probable.   However, in the case of network congestion, the addition of the

redundancy packets makes the degree of network congestion worse, occupies more

bandwidth, and increases the loading of the router so that bursty loss occurs easily.

Motion Picture Experts Group (MPEG) standard is the widely used standard

for video coding and compression.   Thus, MPEG standard is taken as an example as

15   follows.   In MPEG standard, there are three types of video frame: I-frame, P-frame,

and B-frame.   Multimedia information is stored by using a group of pictures (GOP)

as the data storage unit, and a GOP includes fifteen frames, each of which contains

one I-frame, four P-frames, and ten B-frames.

Referring to FIG. 2, it illustrates the relation among different frames of a GOP

20   in MPEG format.   In FIG. 2, X_j^i is an identifier indicating the j-th X type video

frame of the i-th GOP.   For instance, B_2^1 indicates the second B-frame of the first

GOP.   In addition, the arrows among the identifiers indicate which of the frames

(pointed to by an arrowhead) required to be referred to when a frame is being read.

In FIG. 2, data of an I-frame is to be compressed completely with the Joint

Photographic Experts Group (JPEG) format.   Data of each P-frame stores the

difference between the P-frame and either a previous I-frame or P-frame.   Data of

each B-frame stores the differences between the B-frame, either a previous I-frame or

5   P-frame, and either a next I-frame or P-frame.   Therefore, I-frame type is the largest

frame type in data size, and the P-frame type is larger than the B-frame type in data

size, wherein the B-frame type is the smaller frame type in data size.   The display for

an I-frame can be derived from I-frame directly.   As for obtaining the display for a

P-frame, an I-frame is to be referred to.   In addition, for obtaining the display for a

10   B-frame, both an I-frame and a P-frame are to be referred to.   As described above,

the characteristic of the decoding of a frame type that involves referring to other types

of frames is called interframe dependency.   Since MPEG compression format is of

interframe dependency, importance of each frame type is different and the importance

of a frame type is determined by the degree of being referred to.   The more a frame

15   is to be referred to by the other frames, the more important the frame to be referred to

is.   For MPEG format, the sequence of importance of frame types measured from the

most to the less is: I-frame > P-frame > B-frame.

Moreover, since MPEG format has interframe dependency, data of the frames

of higher importance have to be transmitted to the receiving end more completely in

20   the course of network transmission so that the multimedia information can be decoded

and used at the receiving end smoothly.   If data of the I-frame are not completely

received by the receiving end, the multimedia information cannot be decoded at the

receiving end even if the subsequent B-frames and P-frames of the GOP are

transmitted to the receiving end successfully.   Likewise, if data of the P-frames are

25   not received, the relevant B-frames cannot be decoded successfully.

Referring to FIG. 3A, it illustrates a conventional method for determining a

transmission sequence of the packets of a GOP in MPEG format, wherein the GOP

has six packets of I-frame, ten packets of P-frames, and twelve packets of B-frames.

In FIG. 3A, letters i, p, and b denote one I-frame, P-frame, and B-frame packets

5   respectively.   The conventional method is to transmit the packets according to their

importance.   Firstly, the most important I-frame packets are transmitted.   Then, the

less important P-frame packets are transmitted.   Finally, the B-frame packets are

transmitted.

Bursty loss, as described above, occurs in a network that a router discards a set

10   of contiguous packets at a time.   Referring to FIGS. 3B to 3D, they illustrate burty

loss when packets of multimedia information in MPEG format are transmitting by

using the conventional method.   In FIGS. 3B to 3D, a dotted rectangle is used to

encircle the discarded packets when bursty loss occurs.   Turning to FIG. 3B, when

bursty packet loss occurs so that frame packets of less importance, such as the packets

15   302, are discarded, the influence on the receiving of the other frames of the GOP is

insignificant.   However, as shown in FIGS. 3C and 3D, when frame packets of high

importance, such as the packets 304 in FIG. 3C or the packets 306 in FIG. 3D, are

discarded, the other frames of the GOP will not be decoded and used successfully.

In brief, the conventional method has the following disadvantages.

20   1.   When using ARQ technique, it is difficult to determine the

predetermined time optimally since network congestion cannot be predicted.

In addition, it may occur that the response packet sent from the receiving

end is discarded by the router so that the transmission end re-transmits the

packets.

2.   When using FEC technique, a specific number of redundancy

packets are added.   In the case of network congestion, the addition of

redundancy packets would make the degree of network congestion worse

5   and bursty loss occur more easily.

3.   For multimedia information stored and transmitted in a

compression format of interframe dependency, when bursty loss occurs in

frames of higher importance, the other frames, which are received

completely, cannot be decoded and utilized.

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SUMMARY OF THE INVENTION

It is therefore an object of the invention to provide a method for transmitting

multimedia data packets in a network so as to reduce the probability of multimedia

data packets of higher importance to be discarded during transmitting multimedia data

15   packets, and the influence of bursty loss on the quality of service at the receiving end.

According to the object of the invention, it is to provide a method for packet

transmission of multimedia data in a network.   The method is used for transmitting a

group of pictures (GOP), wherein the GOP includes a plurality of frames, each of the

frames includes a plurality of packets.   In addition, the frames are of N types of

20   frame, a first to an N-th type of frame, and a packet of a frame of a type of frame is

called the packet of the type of frame.   The method includes the steps as follows.   (a)

Set i to one, where i is a positive integer.   (b) Form an i-th transmission sequence by arranging the packets of the i-th type of frame.   (c) Form an $(i+1)$-th transmission sequence by putting the packets of the $(i+1)$-th type of frame between packets of the i-th transmission sequence consecutively.   (d) Increment i by one.   (e) Repeat steps

5   (c) to (d) until i is equal to N.   (f) Transmit the N-th transmission sequence.

According to the object of the invention, it is to provide a method for packet transmission of multimedia data in a network.   The method is used for transmitting M groups of pictures (GOPs) including a first GOP to an M-th GOP, where M is an integer greater than one.   Each of the GOPs includes at most N frames, and each of

10   the frames includes at most P packets, where N and P are integers greater than one. The method includes the steps as follows.   (a1) Set i to one, wherein i is a positive integer.   (a2) Set j to one, wherein j is a positive integer.   (a3) It is to determine whether a first type of frame of the j-th GOP has a packet number greater than i; if not, the method proceeds to step (a5).   (a4) Form a first transmission sequence by putting

15   an i-th packet of the first type of frame of the j-th GOP into the first transmission sequence consecutively.   (a5) Increment j by one.   (a6) Repeat steps (a3) to (a5) until j is greater than M.   (a7) Increment i by one.   (a8) Repeat steps (a2) to (a7) until i is greater than P.   (b1) Set i to one, wherein i is a positive integer.   (b2) Set j to one, wherein j is a positive integer.   (b3) Set k to one, wherein k is a positive

20   integer.   (b4) It is to determine whether the k-th GOP has a frame number greater than i; if not, the method proceeds to step (b7).   (b5) It is to determine whether the $(i+1)$-th type of frame of the k-th GOP has a frame number greater than j; if not, the method proceeds to step (b7).   (b6) Put the j-th packet of the $(i+1)$-th type of frame in the k-th GOP between packets of the i-th transmission sequence consecutively.

25   (b7) Increment k by one.   (b8) Repeat steps (b4) to (b7) until k is greater than M.

(b9) Increment j by one.   (b10) Repeat steps (b3) to (b9) until j is greater than P.

(b11) Form an $(i+1)$-th transmission sequence by putting the packets of the frames of

the $(i+1)$-th type of the GOPs between the packets of the i-th transmission sequence

consecutively.   (b12) Increment i by one.   (b13) Repeat steps (b2) to (b12) until i is

5   equal to N.   (b14) Transmit the N-th transmission sequence.

BRIEF DESCRIPTION OF THE DRAWINGS

Other objects, features, and advantages of the invention will become

apparent from the following detailed description of the preferred but non-limiting

10   embodiments.   The description is made with reference to the accompanying

drawings in which:

FIG. 1 (Prior Art) illustrates a conventional network transmission system;

FIG. 2 (Prior Art) illustrates the relation among different frames in a group of

pictures (GOP) in MPEG format;

15   FIG. 3A(Prior Art) illustrates a conventional method for determining a

transmission sequence of the packets of a GOP in MPEG format;

FIG. 3B (Prior Art) illustrates that bursty loss occurs in B-frame packets

during transmission multimedia data packets in MPEG format;

FIG. 3C (Prior Art) illustrates that bursty loss occurs in P-frame packets

20   during transmission multimedia data packets in MPEG format;

FIG. 3D (Prior Art) illustrates that bursty loss occurs in I-frame packets during

transmission multimedia data packets in MPEG format;

FIG. 4 illustrates a network transmission system for determining a

transmission sequence for the packets of a GOP according to a preferred embodiment

5   of the invention;

FIG. 5A illustrates a transmission sequence for the I-frame and P-frame

packets of a GOP according to a preferred embodiment of the invention;

FIG. 5B illustrates a transmission sequence for the I-frame, P-frame, and

B-frame packets of a GOP according to a preferred embodiment of the invention;

10   FIG. 6A illustrates a transmission sequence for the I-frame packets of two

GOPs according to a preferred embodiment of the invention;

FIG. 6B illustrates a transmission sequence for the I-frame and P-frame

packets of two GOPs according to a preferred embodiment of the invention;

FIG. 6C illustrates a transmission sequence for the I-frame, P-frame, and

15   B-frame packets of two GOPs according to a preferred embodiment of the invention;

FIG. 7 illustrates a network transmission system for transmitting the packets of

a GOP according to a preferred embodiment of the invention incorporating with

forward error correction; and

FIGS. 8, 9, and 10 show the results of simulation experiments of comparing

20   the method for transmitting packets in a network according to the invention with the

conventional transmission method respectively.

DESCRIPTION OF THE PREFERRED EMBODIMENT

The principle of the invention is that when packets to be transmitted through a
5   network are of different degree of importance, the packets of higher importance are
spread out and arranged in transmission sequences.   In addition, it is to reduce the
probability of multimedia data packets of higher importance to be discarded during
transmitting multimedia data packets, and the influence of bursty boss on the quality
of service at the receiving end.

10   Referring to FIG. 4, it illustrates a network transmission system for
determining a transmission sequence for the packets of a GOP according to a
preferred embodiment of the invention.   The network transmission system includes a
transmission end 402, and one or more receiving ends such as a receiving end 416.
The receiving end 416 may submit a request for multimedia service to transmission
15   end 402.   On receiving the request from the receiving end 416, transmission end 402
transmits multimedia data which are stored and compressed in the transmitting end in
advance to one or more transmission ends in a way of unicast or multicast in real-time.
The receiving end 416 receives the contiguous data, decompresses them, and play the
decompressed multimedia data in real-time.

20   When receiving the request from the receiving end 416, transmission end 402
divides pre-compressed multimedia data into a series of data packets.   For example,
every frame of a video file in MPEG format is divided into a series of data packets.

Then, the packets are sent to a packet processor 404 to determine transmission

sequences of the packets which are to be transmitted to the receiving end 416.

Since the problem of bursty packet loss in a router may cause users unable to

use the multimedia data received at the receiving end, a method for packet

5   transmission of multimedia data in a network according to the invention is to resolve

this problem and is described as follows.

Referring to FIG. 5A, it illustrates a transmission sequence for the I-frame and

P-frame packets of a GOP according to a preferred embodiment of the invention,

wherein x_i denotes the i-th packet of X frame.   Firstly, the first transmission

10   sequence is formed by arranging packets of I-frames.   Then, packets of secondarily

important P-frames are put between successive packets of the I-frames consecutively.

Since the packet number of the P-frames is greater than the packet number of the

I-frames, when the successive packets of the I-frames have the packets of the

P-frames put between, there are a portion of packets of the P-frames left to be

15   arranged into the transmission sequences.   In this case, the portion of packets of the

P-frames left to be arranged are put between the successive packets of a current

transmission sequence from the header of the current transmission sequence

consecutively.   In FIG. 5A, after packet p_5 is put between packets i_5 and i_6, the

arrangement of packet is started again from the beginning of the transmission

20   sequence to put packet p_6 between packets i_1 and p_1.   By using the method described

above to arrange the other packets of the P-frames, the second transmission sequence

is obtained as shown in FIG. 5A.

Referring to FIG. 5B, it illustrates a transmission sequence for the I-frame,

P-frame, and B-frame packets of a GOP according to a preferred embodiment of the invention. After the determination of the transmission sequence for the I-frame and P-frame packets, the less important B-frame packets are arranged after the last P-frame packet. In this way, packet b_1 is put after the last P-frame packet p_{10} and is

5   put between the packets p_3 and i_4. Then, according to the arrangement for the P-frame packets, the remaining B-frame packets are arranged. By using the method described above to arrange the other packets of the B-frames, the third transmission sequence is obtained as shown in FIG. 5B.

By the method described above, the packet processor 404 determines the

10   transmission sequence of the packets. After that, the packets are sent to an encoder 406 of transmission end 402 according to the third transmission sequence for encoding, and then encoded packets are delivered to the receiving end via the network. When receiving the encoded packets, the receiving end 406 decodes them by using a decoder 412, and puts the decoded packets in original order by using a packet

15   processor 414. Then, the multimedia data can be read and used at the receiving end 416.

By using the invention, two or more GOPs are enabled to be used as a basic unit for the arrangement of transmission sequences of packets. Referring to FIGS. 6A to 6C, they illustrate the determination of transmission sequence of packets of two

20   GOPs as another embodiment of the invention. In addition, each of the two GOPs has six I-frame packets, ten P-frame packets, and twelve B-frame packets. In FIGS. 6A to 6C, characters i, p, and b denote an I-frame packet, a P-frame packet, and a B-frame packet respectively, and x_j^i denotes the j-th packet of X-type frames of the i-th GOP. For instance, b_2^1 denotes the second packet of B-frames of the first GOP.

Referring to FIG. 6A, it illustrates a transmission sequence for the I-frame packets of two GOPs according to a preferred embodiment of the invention. First, a first transmission sequence is formed by arranging I-frame packets of the two GOPs in an alternate manner. The first packet of I-frames of the first GOP (i_1^1) is first put

5 into the transmission sequence, and then the first packet of I-frames of the second GOP (i_1^2) is put into it. Next, the second packet of I-frames of the first GOP (i_2^1) is first put into the transmission sequence, followed by the second packet of I-frames of the second GOP (i_2^2) is put into it. Similarly, the other packets are put into the transmission sequence in the alternate manner. After the last packet of the I-frames

10 of the first GOP (i_6^1) and the last packet of the I-frames of the second GOP (i_6^2) are put into the transmission sequence, the first transmission sequence is obtained as shown in FIG. 6A.

Referring to FIG. 6B, it illustrates a transmission sequence for the I-frame and P-frame packets of two GOPs according to a preferred embodiment of the invention.

15 After the determination of the arrangement of the packets of the I-frames of the two GOPs in the first transmission sequence, a second transmission sequence is formed by putting packets of P-frames of the two GOPs into the first transmission sequence. In the insertion, the I-frame packets of different GOPs are regarded as the same kind of I-frame packets and hence a P-frame packet is put between two adjacent I-frame

20 packets. Since the packet number of the P-frames is greater than the packet number of the I-frames, when packet p_6^1 is put into the first transmission sequence, it is already in the end of the first transmission sequence. In this case, the arrangement of the following packets starts from the leading end of the first transmission sequence. That is, the following P-frame packets are to be put between the successive packets of

25 the leading end of the transmission sequence consecutively. For instance, packet p_6^2

is put between the packets $i_1{}^1$ and $p_1{}^1$. Following the same way of insertion of the P-frame packets for the two GOPs, the second transmission sequence is obtained as shown in FIG. 6B.

5   Referring to FIG. 6C, it illustrates a transmission sequence for the I-frame, P-frame, and B-frame packets of two GOPs according to a preferred embodiment of the invention. After the second transmission sequence is formed by putting all of the P-frame packets into the first transmission sequence, the B-frame packets are to be put into the second transmission sequence sequentially following the same way of insertion of the last P-frame packet so as to form a third transmission sequence. That

10   is, the B-frame packets are put between successive packets of the packets following the last P-frame packet $p_{10}{}^2$ consecutively. For instance, packet $b_1{}^1$ is put between the packets $p_3{}^1$ and $i_3{}^2$. Following the same way of insertion for putting the B-frame packets for the two GOPs into the second transmission sequence, the third transmission sequence is obtained as shown in FIG. 6C.

15   According to the invention, a plurality of GOPs can be used as a basic unit for the arrangement of transmission sequences. When it is required to transmit N GOPs, each frame of data of each GOP is firstly divided into a series of packets before transmission. Secondly, packets of the frames of the first type for all GOPs are arranged so as to form a first transmission sequence. That is, the first packet of the

20   frames of the first type for the first GOP is put into a transmission sequence, and then that for the second GOP is put into the transmission sequence. Similarly, this insertion is repeated for the other GOP in sequence until the first packet of the frames of the first type for the N-th GOP is done. Next, the second packet of the frames of the first type for the first GOP is put into the transmission sequence, and then that for

the second GOP is put into the transmission sequence.   Similarly, this insertion is repeated for the other GOP in sequence until the second packet of the frames of the first type for the N-th GOP is done.   Further, in a similar way, the insertion is repeated for other packets for the N GOPs sequentially, and the last packet of the frames of the first type for the first GOP is put into the transmission sequence, and then that for the second GOP is put into the transmission sequence.   Likewise, this insertion is repeated sequentially until the last packet of the frames of the first type for the N-th GOP is put into the transmission sequence.   In this way, the first transmission sequence is formed.   Then, in order to form the second transmission sequence packets of the frames of the second type for every GOP are put into the first transmission sequence sequentially such that they are put between successive packets of the first transmission sequence consecutively.   After that, packets of the frames of the other type for every GOP are put into the current transmission sequence sequentially to form the next transmission sequence until the N-th transmission sequence is formed by putting packets of the frames of the N-th type for every GOP into the (N-1)-th transmission sequence.   Finally, when transmission the N GOPs, the packets of the GOPs are transmitted according to the N-th transmission sequence.

The method and system provided by the invention can be implemented with data recovery technique, such as ARQ or FEC technique.   If ARQ technique is applied, since the probability and number of I-frames being discarded due to bursty loss are reduced, data and time for retransmission are reduced.   If FEC technique is applied, since the probability and number of packets of I-frames and B-frames, which are of higher importance, being discarded due to bursty loss are reduced, the required redundancy packets are reduced.   In this way, the problem that a plenty of redundancy packets make the network congestion worse can be avoided.   Moreover,

since FEC technique does not involve packet retransmission, it is more appropriate for use in transmission of multimedia data in real-time.   Therefore, when transmitting multimedia data through the network, FEC technique is preferred to be used in the implementation of the invention.

5   Referring to FIG. 7, it illustrates a network transmission system for transmitting the packets of a GOP according to a preferred embodiment of the invention incorporating with forward error correction.   In FIG. 7, the difference between FIG. 2 and FIG. 7 is that an encoder 706 of a transmitting end 702 in FIG. 7 employs FEC technique to encode packets.   In addition, a packet permutation

10   module is coupled to the encoder 706 to add redundancy packets to the encoded packets.   The redundancy packets and the packets in predetermined order are transmitted as a whole.   On the other hand, a packet re-permutation module 714, coupled to a decoder 716 of a receiving end 720, is used to perform the reverse operation so as to separate the redundancy packets, wherein the decoder 716 performs

15   decoding by using FEC technique.

        Referring to FIGS. 8, 9, and 10, they show the results of comparing the method according to the invention with the conventional transmission method by using computer simulation.   In simulation, every GOP has fifteen frames and includes one I-frame, four P-frames, and ten B-frames.   In addition, the I-frame,

20   P-frame, and B-frame have 18750, 6250, and 2500 bytes respectively, and the maximum transfer unit (MTU) is 512 bytes.   In this simulation experiment, Gilbert model is employed to model the characteristic bursty loss in the internet.   The purpose of FIGS. 8 to 10 is to show the difference of the number of redundancy packets needed to be added to the packets between the method according to the

invention combining with FEC technique and the convention transmission method

combining with FEC technique when both methods are used to transmit the

multimedia data in MPEG format with identical quality of service (QoS).   The

definition of QoS is to indicate the percentage of I-frame packet not lost in a GOB.

5   Referring to FIG. 8, it illustrates the simulation experiment results of

comparing the method for transmitting packets in a network according to the

invention with the conventional transmission method.   In the simulation, the loss rate

is set to 5% and QoS is set to 95%, that is, the probability of receiving complete

I-frame packets in the receiving end that receives the GOPs is 95%.   When

10   multimedia data in MPEG format are transmitted under the conditions, it is to obtain

reduction ratios of the number of redundancy packets needed to be added to the

packets by using the method according to the invention combining with FEC

technique to that of the conventional transmission method combining with FEC

technique.   It can be observed from FIG. 8, when bursty packet loss occurs, the

15   reduction ratio ranges between 27% to 43%.   In addition, as the number of lost

packet increases, the number of redundancy packets according to the invention is

reduced as compared with the conventional method.

Referring to FIG. 9, it illustrates the simulation experiment results of

comparing the method for transmitting packets in a network according to the

20   invention with the conventional transmission method.   In the simulation, the loss rate

is set to 5%, QoS is set to 95%, and average number of packets dropped in a bursty

loss is set to three packets.   When multimedia data in MPEG format are transmitted

under the conditions, it is to obtain reduction ratios of the number of redundancy

packets needed to be added to the packets by using the method according to the

invention combining with FEC technique to that of the conventional transmission

method combining with FEC technique.   As observed from FIG. 9, when bursty

packet loss occurs, the reduction ratio ranges between 36% to 43%.   In addition, as

the number of GOP increases, the number of redundancy packets according to the

5   invention is reduced as compared with the conventional method.

Referring to FIG. 10, it illustrates the simulation experiment results of

comparing the method for transmitting packets in a network according to the

invention with the conventional transmission method.   In the simulation, the loss rate

is set to 5%, QoS is set to 95%, average number of packets dropped in a bursty loss is

10   set to three packets, and two GOPs are required to be transmitted.   When multimedia

data in MPEG format are transmitted under the conditions, as the probability of the

occurrence of bursty loss increases, the number of redundancy packets needed to be

added to the packets by using the method according to the invention combining with

FEC technique is reduced as compared with that of the conventional transmission

15   method combining with FEC technique.   As observed from FIG. 10, when bursty

packet loss occurs, the reduction ratio ranges between 32% to 43%.   Although, as the

probability of the occurrence of bursty loss increases, it is required more redundancy

packets according to the invention for maintaining identical QoS, the number of

redundancy packets according to the invention is much less than that of the

20   conventional method.

In the above embodiment, the multimedia data in MPEG format are

transmitted in the Internet for the illustration of the invention.   In addition to

transmission of the multimedia data in MPEG format in the Internet, the invention can

be applied to the transmission of continuous multimedia data in other interframe

dependent format in a network where bursty packet loss phenomenon occurs.

As described above, the method for packet transmission of multimedia data in a network according to the invention has the following advantages.

1. Reducing the influence of bursty packet boss on the quality of service at the receiving end.   By the permutation of the sequence for transmission according to the invention, the data packets of higher importance are transmitted distributively through the transmission sequence.   Hence, when bursty loss occurs due to network congestion, the number of discarded data packets of higher importance is reduced. In this way, the influence of bursty boss on the quality of service at the receiving end is reduced.

2. Resolving the disadvantage of the conventional transmission technique. The method and system provided by the invention can be implemented with data recovery technique, such as ARQ or FEC technique.   If ARQ technique is applied, data and time for retransmission are reduced.   If FEC technique is applied, the required redundancy packets are reduced.   In this way, the problem that a plenty of redundancy packets make the network congestion worse can be avoided.

3. After the permutation of the frame packets of multiple GOPs according to invention, frame packets of higher importance are spaced apart from one another.   In this way, the influence of bursty packet loss is reduced more much.

While the invention has been described by way of example and in terms of the preferred embodiment, it is to be understood that the invention is not limited to the disclosed embodiment.   To the contrary, it is intended to cover various modifications

and similar arrangements and procedures, and the scope of the appended claims

therefore should be accorded the broadest interpretation so as to encompass all such

modifications and similar arrangements and procedures.